PROCESS AND DEVICE FOR ESTIMATING THE IMPULSE RESPONSE OF AN INFORMATION TRANSMISSION CHANNEL, IN PARTICULAR FOR A CELLULAR MOBILE TELEPHONE

Field of the Invention

The invention relates to the transmission of digital information, and more particularly but not exclusively, within a cellular mobile telephone domain

5 according to the GSM (Global System for Mobile Communication) standard.

The invention relates more particularly to the estimation of the impulse response of a transmission channel conveying information from a

10 sender to a receiver via a means of propagation. The means of propagation may be air for cellular mobile telephones, or any other means of propagation, such as a cable in other applications, for example.

Background of the Invention

15 A fundamental factor limiting the performance of a digital communication system is the phenomenon known as inter-symbol interference, which is well known by a person skilled in the art. Inter-symbol interference causes at the receiver level temporal

20 occupation of each symbol or bit transmitted, which is

longer than the initial duration of the symbol. This is also referred to as the bit time, for example.

Stated otherwise, the signal received at a given instant does not depend on one symbol or bit alone, but also on the other symbols or bits sent, which extends over durations greater than those of one symbol or bit time. In practice, the signal received at a given instant depends on the symbol concerned, and also essentially on adjacent symbols.

The causes of inter-symbol interference are multifold. One of them is due in particular to the multiple propagations of the signal between the sender and the receiver when the signal is reflected or diffracted by various obstacles. This leads to the reception of several signal copies that are mutually shifted temporally. Moreover, this interference between symbols is produced not only by the propagation between the sender and the receiver, but also by the sending/receiving devices themselves, such as the modulator, filter, etc.

During communications with interference between symbols, the problem arises on estimating the impulse response of the transmission channel. The quality of this estimate depends on the capacity to eliminate the interference between symbols, and hence to take corrective action regarding symbols sent.

Generally, the estimate of the impulse response of the channel, or more simply the channel estimate, is performed within the GSM telephone domain by using least squares techniques, and by using a predetermined sequence of symbols which is known to the sender and to the receiver. This is referred to by the term training sequence, which is well known by a person

skilled in the art. This training sequence is present within each symbol train or burst sent. When the characteristics of the channel are sufficiently well estimated, the estimated coefficients of the impulse

5 response of the channel are used in an equalization processing operation, also well known by the person skilled in the art. This is done to decipher the signal received. That is to say, the logic values of the symbols or data sent in the train are retrieved.

10   The equalization processing operation is conventionally followed by the channel decoding processing operations intended for correcting any errors. Channel decoding is followed by another decoding, known as source decoding, which is intended

15 for reconstructing the information (speech, for example) initially coded at the level of the sender.

The article by Khayrallah et al., entitled "Improved Channel Estimation With Side Information" (1997, IEEE) and the corresponding U.S. Patent

20 No. 5,838,739, discloses a method of channel estimation. This estimation is based on the combined use of training sequences, and of the known characteristics of the sending and receiving means (filters), and includes determining the estimated

25 coefficients of the impulse response of the channel.

However, such a method has drawbacks. One of them resides in the fact that this method depends on the information transmitted, namely the training sequences, which may be different depending on the

30 systems used. Consequently, it is necessary to store in the portable telephone, for example, as many matrices as there are different possible training sequences. Another drawback resides in the fact that

such a method is not applicable to so-called blind channel estimates, that is to say, when training sequences are not available.

Summary of the Invention

5   In view of the foregoing background, an object of the present invention is to provide an improvement to channel estimation which can be applied equally well to learned estimates (i.e., those with training sequences) and blind estimates (i.e., those 10 with no training sequences).

Another object of the present invention to provide an improvement of this kind which is extremely straightforward to implement.

The invention therefore provides a process 15 for estimating the impulse response of an information transmission channel comprising an information sender, an information receiver and a means of propagation, such as air, for example, extending between the sender and the receiver.

20   According to a general characteristic of the invention, a first estimate is made of the impulse response of the channel considered as a whole, then this first estimate is corrected, independently of the information transmitted, by taking account of the fact 25 that the impulse response of the sender and the impulse response of the receiver are known. This is done to obtain a corrected final estimate of the impulse response of the channel.

Stated otherwise, the invention proceeds in 30 two phases. First, the channel is estimated as a whole, including the channel that is in association with the sender, the receiver and the means of

propagation, but without taking account the fact that the impulse response of the sender and of the receiver are known. Stated otherwise, this first estimate is made in a conventional manner, by any known method,

5 such as learned or blind. Next, in a second phase, this first estimate is corrected by employing the known elements, namely the impulse responses of the sender and of the receiver.

Apart from the fact that the process

10 according to the invention is perfectly compatible with blind or learned channel estimates, it is, with regards to the correction phase, totally independent of the data sent. This is so since it involves only the intrinsic and known characteristics of the sending and

15 receiving means, such as the modulators, demodulators and various filters, for example.

According to one mode of implementation of the process according to the invention, the coefficients of the final estimate of the impulse

20 response of the channel are obtained by multiplying the coefficients of the first estimate of the impulse response of the channel by a matrix whose coefficients are predetermined and representative of the impulse responses of the sender and of the receiver, and are

25 independent of the information transmitted on the channel.

Thus, the invention is noteworthy in the sense that it requires the storage of just a single transformation matrix, whose coefficients in particular

30 are independent of the data transmitted, and consequently independent of the training sequences where these exist.

Another aspect of the invention is directed to a device for estimating the impulse response of an information transmission channel comprising an information sender, an information receiver and a means

5 of propagation extending between the sender and the receiver.

According to a general characteristic of the invention, the device comprises first estimation means able to make a first estimate of the impulse response

10 of the channel considered as a whole, and second estimation means connected to the first estimation means for correcting this first estimate, independently of the information transmitted for delivering a corrected final estimate of the impulse response of the

15 channel. This is done by taking account of the fact that the impulse response of the sender and the impulse response of the receiver are known.

According to one embodiment of the invention, the second estimation means comprises a memory

20 containing a matrix whose coefficients are predetermined and representative of the impulse responses of the sender and of the receiver, and are independent of the information transmitted on the channel. The second estimation means also comprises

25 processing means able to multiply the coefficients of the first estimate of the impulse response of the channel by the matrix to obtain the coefficients of the final estimate of the impulse response of the channel. The first estimation means are able to make a first

30 blind estimate, or else a first learned estimate.

Another aspect of the invention is directed to a cellular mobile telephone incorporating a device

for estimating the impulse response of an information transmission channel as defined above.

Yet another aspect of the invention provides a computer program comprising program-code means implementing the estimation process, as defined above, when the program is executed within a processor.

The invention further provides a storage device, for example, a read-only memory, capable of being read by a processor, and containing program-code means able to implement the estimation process as defined above when the program is executed within the processor.

Brief Description of the Drawings

Other advantages and characteristics of the invention will become apparent on examining the detailed description of embodiments and modes of implementation that are in no way limiting, and the appended drawings, in which:

Figure 1 is a block diagram illustrating the basic parts of a sender and of a receiver for implementation of the process according to the present invention;

Figure 2 is a block diagram illustrating in greater detail part of the receiver illustrated in Figure 1;

Figure 3 is a block diagram illustrating the components of the impulse response of a transmission channel according to the present invention; and

Figure 4 is a simplified flowchart of a mode of implementation of the process according to the present invention.

Detailed Description of the Preferred Embodiments

It is now assumed, referring more particularly to Figure 1, that the invention is applied to the domain of cellular mobile telephones, for

5   example, such as those of the GSM network, for example. In Figure 1, the reference EM designates a sender or transmitter comprising upstream a coding block TCC for receiving the useful data to be transmitted, such as speech for example, and performing in particular

10 channel coding conventional processing for introducing redundancies into the data stream. The output from the TCC block includes binary information blocks.

The TCC block is conventionally followed by a modulator performing, for example, a quadrature

15 modulation of the QPSK or 8PSK type, according to terminology well known to a person skilled in the art, and transforming the binary signal into an analog signal. This analog signal is then filtered in a send filter FE before being sent towards the receiver via an

20 antenna ANT1. The means of propagation MPR between a sender EM and a receiver TP, here including a cellular mobile telephone, is air in the present case.

The receiver TP, or cellular mobile telephone, comprises at the front end an antenna ANT2

25 coupled to an analog stage PAN, carrying out a frequency conversion to bring the modulated signal received into the baseband, and a filtering to retain only the useful part of the spectrum. After sampling and analog/digital conversion in a converter CAN, the

30 role of the digital stage is to produce an estimate of the transmission channel. This is done in block BST, wherein a more detailed structure is provided in Figure 2 which will be described below. This is done to

eliminate the inter-symbol interference by virtue of this estimate, i.e., by an equalization performed in a block BEQ, and in general, to perform error correction. This includes conventional channel decoding (block TDC) based, for example, on a Viterbi decoder, etc.

Within the meaning of the present invention, the transmission channel is formed by the elements situated upstream of the channel estimator. In particular, this includes the analog sending and receiving devices as well as the physical means of propagation MPR. It should be noted here that it is also possible to find and to take into account digital processing operations (filtering, for example) performed upstream of the channel estimator but downstream of the analog reception stage.

The model or impulse response H(z) of the transmission channel to be estimated is illustrated diagrammatically in Figure 3. The channel impulse response RICT is in fact the product of the impulse response RIE of the sender, of the impulse response RIP of the means of propagation, and of the impulse response RIR of the receiver.

More precisely, H(z) is defined by formula (I) below:

$$H(z) = G_e(z) * C(z) * G_r(z) \qquad (I)$$

or, more simply, H(z) is equal to:

$$H(z) = G(z) * C(z) \qquad (II)$$

in which G(z) represents the impulse response of the sending and receiving devices, and C(z) is the impulse response of the physical means of propagation.

In the present case, G(z) and C(z) are

5   polynomials in z^{-1}. The number of complex coefficients is predetermined and depends in particular on the characteristics of the sending and receiving devices, and also on the signal propagation environment (hilly, urban or other path). The number of complex

10   coefficients is generally based on a worst case. Thus, from a propagation point of view, the mountainous environment is effectively a worst case as compared with the urban or rural environment, for example.

It is assumed, by way of example, that the

15   impulse response H of the transmission channel taken as a whole is a polynomial in z^{-1} of degree 7 with eight complex coefficients h_0-h_7, which we seek to estimate. H(z) is therefore defined by formula (III) below:

$$H(z) = h_0 + h_1 z^{-1} + \ldots\ldots + h_7 z^{-7} \quad (III)$$

20   A first estimate 40 of the coefficients h_0-h_7 is first made (Figure 4). This first estimate is made in a conventional manner. More precisely, if this first estimate is a learned estimate, that is to say, one which uses training sequences formed of a

25   succession of bits of known value, each received signal sequence R is defined by the matrix formula (IV) below:

$$R = S*H + N \quad\quad\quad (IV)$$

in which S designates the matrix representative of the training sequence, H represents the vector formed of

the coefficients h_0-h_7 and N an additional vector
representative, in particular, of interference and of
thermal noise.

A conventional way of solving this system,
5 that is to say, of estimating the coefficient h_i of the
impulse response H of the channel taken as a whole,
includes using a least squares method which determines
H as being the vector which minimizes the (Euclidean)
norm of the vector N. The person skilled in the art is
10 then aware that the vector H is defined by the formula
(V) below:

$$H = (S*S)^{-1} S* R \qquad\qquad (V)$$

in which the notation "*" designates the transposed
complex conjugate matrix.
15 Of course, other methods are also usable to
estimate the coefficients h_i of the impulse response of
the channel taken as a whole. These other methods are
also well known to the person skilled in the art and
they will not be discussed in further detail here.
20 It is also possible to use so-called blind
estimates, that is to say, ones in which no
predetermined training sequences are used. The person
skilled in the art is also aware of various methods
making it possible to perform blind estimations of the
25 impulse response H of a channel taken as a whole. By
way of indication, we may cite the article by Jitendra
K. Tugnait, entitled "Blind Estimation Of Digital
Communication Channel Impulse Response", IEEE
Transactions On Communications, Vol. 42, No. 2/3/4,
30 February/March/April 1994.

The channel estimate is made in a block BST
(Figure 2). More precisely, the first estimate is made
in a subblock BST1 of the block BST. The entire block
BST may be embodied in hardware, for example, as a

5   signal processing processor, with the processing
operations performed in the block BST1 being carried
out in software. These processing operations are
therefore in the form of program codes which can easily
be written by the person skilled in the art on the

10   basis of the functional definition of these processing
operations. The program-code is then, for example,
stored in a read-only memory associated with the
processor. Consequently, an entirely hardware
embodiment of the block BST is also possible, for

15   example, in the form of an application specific
integrated circuit (ASIC).

Once this first estimate has been obtained,
it will be corrected in a block BCR, which is also
functionally embodied in software within the signal

20   processing processor. It is again assumed by way of
example that the impulse response C of the physical
propagation means is a polynomial with five complex
coefficients of degree 4, defined according to formula
(VI) below:

25           $C(z) = c_0 + c_1 z^{-1} + c_2 z^{-2} + c_3 z^{-3} + c_4 z^{-4}$ (VI)

Likewise, it is assumed by way of example
that the impulse response G of the sending and
receiving devices is a polynomial of degree 3 with four
complex coefficients, defined by formula (VII) below:

30           $G(z) = g_0 + g_1 z^{-1} + g_2 z^{-2} + g_3 z^{-3}$ (VII)

The convolution product (polynomial product)
$H(z) = G(z) * C(z)$ is then represented in matrix fashion
by the equation $H = G * C$, in which H is the column
vector formed of the coefficients h_i, C is the column

5 vector formed of the coefficients c_i, and G is the
matrix (here an 8 x 5 matrix) defined by formula (VIII)
below:

$$G = \begin{pmatrix} g_0 & 0 & 0 & 0 & 0 \\ g_1 & g_0 & 0 & 0 & 0 \\ g_2 & g_1 & g_0 & 0 & 0 \\ g_3 & g_2 & g_1 & g_0 & 0 \\ 0 & g_3 & g_2 & g_1 & g_0 \\ 0 & 0 & g_3 & g_2 & g_1 \\ 0 & 0 & 0 & g_3 & g_2 \\ 0 & 0 & 0 & 0 & g_3 \end{pmatrix} \qquad \text{(VIII)}$$

More generally, since the impulse response

10 $H(z)$ is a linear combination of $G(z)$ and its delayed
versions, the number of columns of the matrix G is
equal to the number of coefficients of the impulse
response $C(z)$. Each column of the matrix G comprises
the coefficients g_i supplemented with 0s. The

15 coefficients of a column are shifted by one row with
respect to the coefficients of the previous column.

The corrected final estimated impulse
response of the channel, referenced HF, must then be
written according to formula (IX):

20 $$HF = GC \qquad \text{(IX)}$$

and must also be as close as possible to the impulse
response H. Stated otherwise, HF must minimize the
criterion defined by formula (X) below:

$$\|HF - H\|^2 \qquad\qquad (X)$$

5   It can then be shown that the estimated final
impulse response of the channel is defined by formula
(XI) below:

$$\qquad\qquad\qquad\qquad (XI)$$
$$HF = \underbrace{G\,(G*G)^{-1}G*H}_{M}$$

The predetermined matrix M in the present
10   case is a matrix of size 8 x 8 whose coefficients take
account only of the coefficients g_i and are independent
of the data transmitted. This matrix M is stored in a
memory MM of the block BST. The block BCR therefore
performs a correction 41 (Figure 4) of the coefficients
15   h_i of the impulse response H of the channel. This is
done to deliver the coefficients hf_i of the corrected
impulse response HF of the channel. It is these
coefficients hf which will be used in the equalization
block BEQ of the telephone TP (Figure 1).
20   The equalization block will solve the
equation defined by the matrix formula (XII) below:

$$R = b*HF + N \qquad\qquad (XII)$$

in which R designates the vector representative of the
sequence of bits received, b designates the vector
25   representative of the useful bits of the pulse train to
be decoded, and HF designates the convolution matrix in
which each column comprises the coefficients hf_i

supplemented with 0s. The coefficients of a column are shifted by one row with respect to the coefficients of the previous column, and N designates the noise vector. The manner of solving this equation (equalization) is

5 well known to the person skilled in the art. In this regard, there are numerous equalization algorithms.

Among these equalization processing operations, two major classes may be considered. A first class is those which carry out detection symbol

10 by symbol, such as for example, the algorithm known to the person skilled in the art by the term DFE (Decision Feedback Equalization) and whose essential aspects are, for example, described in the work by John G. Proakis, entitled "Digital Communications", third edition,

15 McGraw-Hill, Inc.

A second class is those which carry out the detection of a sequence of symbols, such as for example, the algorithm known by the term MLSE (Maximum-Likelihood Sequence Estimation) or that known by the

20 term DFSE (Decision Feedback Sequence Estimation), also well known to the person skilled in the art. These two algorithms have formed the subject of numerous publications. The person skilled in the art will in particular be able to refer, with regards to the MLSE

25 algorithm, to the work by John G. Proakis cited above, and with regards to the DFSE algorithm, to the article by Hans C. Guren and Nils Holte, entitled "Decision Feedback Sequence Estimation for Continuous Phase Modulation on a Linear Multipath Channel", IEEE

30 Transactions on Communications, Vol. 41, No. 2, February 1993.

Symbol by symbol detection algorithms have very low complexity relative to sequence-based

detection algorithms, but give inferior performance. This is why equalization algorithms using sequence-based estimation are generally chosen in preference. The bits b thus delivered are then decoded in the

5 channel decoding block TDC. The equalization algorithms and channel decoding algorithms are, for example, also executed by the signal processing processor.

The invention is not limited to the

10 embodiments and modes of implementation described above, but embraces all variations thereof. Thus, in the example described, H is a linear combination of G and of its delayed versions. Of course, however, it is possible to consider half-integer delays, or even finer

15 delays, by using an oversampling of the signal.

Moreover, the estimation device according to the invention can be incorporated not only into a cellular mobile telephone, but also into the reception pathway of a base station or more generally into any

20 digital information receiver.